emagic

evoc 20

Studio Tools

>> Version 1.0, January 2002

>> English Edition

User Manual

Mac

Microsoft Windows Compatible

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1 Welcome …

… and thank you for your purchase of the 20 Band Emagic Vocoder—EVOC 20. We are proud of this collection of three excellent plug-ins for your Logic production environment. We are confident that you will find them an extremely versatile addition to the extensive array of effects processing options included with your version of Logic.

This manual will introduce you to the concept and functionality of the three plug-ins in the EVOC 20 package. Please read it thoroughly to make the most of your new Logic effects.

The EVOC 20 package consists of two Vocoders and one Filter Bank. The Vocoders are modelled after the finest analog vocoders.

What is a Vocoder? Put simply, a vocoder transfers the sonic characteristics of the signal arriving at the *analysis* input to the signal arriving at the *synthesis* input. The classic "vocoder" sound uses speech as the analysis signal and a synthesizer sound as the synthesis signal. This classic sound was popularized in the late 70's and early 80's. You'll probably know it from tracks such as; "O Superman" by Laurie Anderson, "Funky Town" by Lipps Inc. and numerous Kraftwerk pieces—from "Autobahn" and "Europe Endless" up to "The Robots" and "Computer World".

Away from these "singing robot" sounds, vocoding has also been used in many films. As examples; the Cylons in Battlestar Galactica, and most famously, on the voice of Darth Vader from the Star Wars saga.

Vocoding, as a process, is not strictly limited to vocal performances. You could use a drum loop as the analysis signal to shape a string ensemble sound arriving at the synthesis input.

To perform these different tasks, the EVOC 20 features three discrete processors, each with unique capabilities. These are the:

- EVOC 20 PS—a vocoder with a built-in polyphonic synthesizer sound engine (an Audio Instrument).
- EVOC 20 TO—a monophonic pitch tracking vocoder (plug-in).
- EVOC 20 FB—a formant filter bank (plug-in).

The synthesis signal of the EVOC 20 PS is provided by its built-in 16-voice polyphonic synthesizer—hence the “PS” in the name. You can play this polyphonic synthesizer in realtime via MIDI. The output signal of the synthesizer is shaped by the signal arriving at the analysis input, producing “classic” vocoded choirs.

The EVOC 20 TO (the “TO” stands for Tracking Oscillator), on the other hand, derives its synthesis signal from a monophonic oscillator. This synthesizer follows the pitch of incoming audio data arriving at the analysis input. It works best with monophonic signals. Input of non-vocal material, such as drums, leads to extremely interesting sonic results. Alternatively to the Tracking Oscillator, the EVOC 20 TO can use a freely selectable audio signal as the synthesis signal.

The EVOC 20 FB consists of the two formant filter banks—the heart of any Vocoder. Each bank features independent volume faders for each band, allowing levels to be set freely—ranging from “unchanged” through to “silence”. The latter completely suppresses the selected formants in the overall sound spectrum. Use of the *Formant Stretch* and *Formant Shift* parameters provide total control over the position and width of the filter bands. In addition, you can also crossfade between the two filter banks.

As with all of Logic’s effects, any adjustments to the EVOC 20’s controls can be recorded and played back in real-time. Such parameter automation data can be created and edited using any suitable MIDI editor window within Logic.

We wish you many years of inspiration, work, fun and productivity with the EVOC 20!

Your EMAGIC Team

2 Installation

What the Package Includes

Your EVOC 20 package contains the following:

- The Emagic *Software CD* with the current Logic version. There's also a collection of settings (sounds) for the EVOC 20.
- This user manual.
- The sealed *Registration Return Envelope*. Please do not open it until you have read the following paragraph. The sealed *Registration Return Envelope* contains the *International Registration Card*. Attached to the card, you will find a barcode sticker with a number on it.

Sealed Envelope and Registration Card

The envelope is sealed. The act of opening the envelope indicates your agreement with, and acceptance of, our licensing conditions and terms of trade.

Please open the envelope carefully along the seal. Do not tear the envelope. It can be resealed, and reused for registration, by enclosing the *International Registration Card*, and posting it, if you wish to register by mail.

A barcode sticker with a number is attached to this card. This number is a temporary authorization code. It must be typed into the *XSKey Authorization window*, as described in *The XSKey Authorization Window* chapter, from page 12 onwards.

After you have entered this authorization code, you may use the EVOC 20 for a period of one month, with no functional restrictions. During this time (preferably right now), you will need to register your product. After registration, you will receive a further authorization code, for unlimited use of the EVOC 20. This "unlimited use" authorization code must also be entered in the *XSKey Authorization window*.

Registering Online

If you have Internet access, please register the EVOC 20 online. This is the simplest and fastest method.

Keep the *International Registration Card* and the serial number of your XSKey handy. You'll need both to register your EVOC 20.

You can find the XSKey serial number on the barcode stickers that came with your Logic 5 version, and under **Help > XSKey Authorization** (Windows) or > **XSKey Authorization** (Mac OS).

Start your web browser and navigate to:

www.emagic.de/registration

- Input the requested data.
- A successful online registration will be indicated by an onscreen confirmation message and via e-mail.
- After a short processing period, you will receive your authorization code for unlimited use via e-mail.

Registering by Mail

If you don't have Internet access, you may register by mail.

- Please enter the appropriate details on the *International Registration Card* carefully, and completely.
- Attach one of the XSKey serial number stickers (as supplied with your Logic package) onto the respective field of the card.
- Insert the *International Registration Card* into the *Registration Return Envelope*.

Please allow for a period of 10 working days to process the card. You will receive the authorization code for unlimited use by mail. The online registration method is preferable, and more convenient.

Updates and Support

The newest software versions are available for download from our website. Should you encounter any difficulties, we also offer a free online support center. You may also consult one of our technicians by telephone.

- Visit www.emagic.de
- Support via our Hotline:
  In the USA: e-mail: support@emagicusa.com
  phone 1-530-477 1050, fax 1-530-477 1052
  In Germany: e-mail: support@emagic.de
  phone +49-(0)4101-495-110
- The InfoWeb is an almost inexhaustible resource for all Emagic products. Its clear layout offers instant access to up-to-date insider information, answers to compatibility issues and troubleshooting help. This is the URL:

http://www.emagic.de/english/support/infoweb/

3 Getting Started

The EVOC 20 is integrated within the Logic software. To make use of the EVOC 20, an installed copy of Logic Audio, Gold, Platinum or MicroLogic AV 5 (or higher) is required.

Please start the Logic 5 installer located on the Emagic Software CD which ships with the EVOC 20. Follow the on-screen installer instructions. The installer updates the EVOC 20 and Logic 5 software components if necessary.

Your stored Plug-In settings will not be erased by this process.

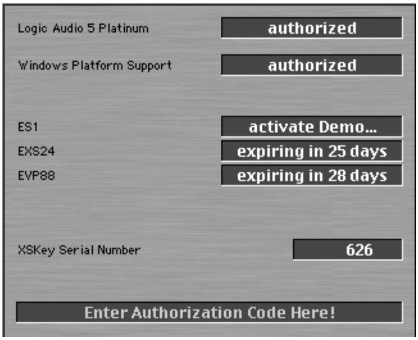
If your Logic 5 series program (i. e. MicroLogic AV, Logic Audio, Logic Gold, Logic Platinum) is not available as an installer option, the most up-to-date version is already installed on your hard drive.

Depending on the volume number of the Emagic Software CD, the installer offers the option to also install other programs of Logic 5 series. These can then be tested by switching the XSKey to the respective demo setting.

The EVOC 20 is copy protected and authorized via the XSKey (Expandable System Key). This is how it works:

The XSKey Authorization Window

Open the *XSKey Authorization window* by selecting **Help > XSKey Authorization** (Windows) or > **XSKey Authorization** (Mac OS). The window shows the authorization status for all *available* software instruments and add-on modules. The authorization code for each is stored in the XSKey. Please take good care of your XSKey!



The window also shows the serial number of your XSKey. All codes, for all products, are entered in the "*Enter Authorization Code Here!*" field. Click once on the field to enter a code. The following describes the messages you may see in the *XSKey Authorization window.*

authorized:

The module is purchased, authorized and ready for "unlimited" use.

expiring in … days:

This module is in a fully functional demo period for the specified number of days. Purchase, and registration, with Emagic will provide you with a code to permanently authorize the module. If no code is supplied within the time period, the demo will expire.

It is recommended that you do *not* attempt to change the date of the system clock during an active demo period, as this may reduce the time before the demo expires.

activate Demo…

The module is not active, but it is possible to enable it's demo mode. To do so, click once in the desired "Activate Demo…" field. Please note that the first time it is started, the demo mode can not be stopped, and will continue to count down! If a

permanent license/authorization code is not purchased within the demo period, use of the module will expire.

expired

The demo period is over. It is not possible to use the module until a valid license code is entered.

empty field

The module is not active, and no demo mode is available. The only way to activate such modules is by entering a license code.

The "Audio Instrument" Object Type

The EVOC20 PS is a software instrument with an integrated sound synthesis engine, which can accept MIDI note input. As opposed to the EVOC20 PS, the EVOC20 TO and EVOC20 FB are effect plug-ins for the inserts of audio objects.

Routing possibilities for the EVOC20 are determined by the version of Logic used. The number of EVOC20 instances which can be run simultaneously is dependent on the availability of computer processing resources, and also on the version of Logic used.

The EVOC20 PS can be found in the hierarchical menu of the top insert slot of any Audio Instrument channel, once the authorization or demo mode is engaged. The following explanations are relevant for Audio Instruments like the EVOC20 PS. After these explanations, we will do a quick tour on the EVOC20 TO and EVOC20 FB.

The default song—the song that opens automatically if you move your Autoload song away from the Logic program folder—features a number of pre-configured Audio Instruments.

An Audio Instrument is an audio object (an Audio Track in MicroLogic AV) with the **Cha** parameter switched to one of the **Instruments** (**1—32**, dependent on Logic version). Any audio

object can operate as an Audio Instrument by changing the **Cha** parameter in the Object Parameter box. Audio objects are created in the Environment by selecting **New > Audio Object.** To create a new Audio Instrument in MicroLogic AV, you can simply select **Track > Create Audio Instrument**.

You can only insert the EVOC 20 PS plug-in into the *top* slot of an Audio Instrument channel.

Important!

Using the EVOC20 PS

The EVOC 20 PS differs from the other two EVOC 20 effects in that it can accept MIDI note input. This ability allows you to “play” the polyphonic sound engine of the EVOC 20 PS. The signal of this sound source will be shaped by the audio track you selected as a “**side chain**”.

The “Side Chain” principle (or “Keying”) was initially introduced in analog dynamics processors. The “Side Chain” (or “Key Input”) is a control input for a signal, which is not heard directly, but has a direct influence on the output signal.

Please follow these steps to make use of the EVOC 20 PS:

- Following the installation of the EVOC 20, please start Logic.
- Select or create a new Audio track in the Arrange window.
- Insert (or record) an audio file—we’ll take a vocal part to start with—onto this Audio track by pressing ⇧ while clicking onto the arrange area with the pencil tool to the right of the track name in the Track List.
- This will launch a file browser, allowing you to select the desired audio file. Click once on the audio file you wish to use and press **Open**. The file will be inserted at the selected location.
  It may be worthwhile setting up a cycle region in the Arrange window, allowing you to continually cycle the audio part. This will make experimentation easier.

  Tip
- Select or create a new Audio Instrument track in the Arrange window. By selecting this track, the Audio Instrument will

be activated, enabling it to receive MIDI data from your keyboard.

- Open the Mixer window, or a screenset with an opened mixer, respectively, or the Environments Audio layer, if you're running Logic Gold or Platinum. You may use the Windows menu or a Key Command (: ⌘M/PC: ctrl M), or you can simply double-click on the Audio Instrument track name to launch the Audio layer of the Environment window.
- Click-hold on the top plug-in slot of the Audio Instrument channel corresponding to the selected Arrange window track, and a hierarchical menu will open.
- Browse to the **EVOC 20 PS** entry in the **Stereo > Logic** group of the plug-in list. Once selected (highlighted), release the mouse button.
- This will launch the plug-in window.
- In the gray area at the top of the plug-in window, click-hold on the **Side Chain** flip menu, and select the Audio track that contains the audio file.
- Ensure that the corresponding Audio Instrument track is selected in the Arrange window.
- The EVOC 20 is now ready to accept incoming MIDI data, and has been assigned to see the output from the selected audio track via a side chain.
- In the Track Mixer or Environment Audio layer (not the Arrange!), mute the audio track (the vocal track) serving as sidechain input.
- Press the play button on the Transport Bar, or use the *Play* Key Command (0 on the numeric keypad).
- And now … as the audio file is playing back, play your MIDI keyboard. You may record your MIDI performances as with other MIDI or Audio Instrument tracks.
- In the Track Mixer (or Environment Audio layer), adjust the volume levels of the EVOC 20 PS and the audio track used for the *Side Chain* to your taste.
- Do a little experimentation with the knobs, sliders and other controls. Have fun, and feel free to insert further effect plug-ins on the channel or busses to further enhance the sound.

Using the EVOC20 TO and EVOC20 FB

- Following the installation of the EVOC20, please start Logic.
- Select or create a new Audio track in the Arrange window.
- Insert (or record) an audio file—preferably a vocal part—onto this Audio track by pressing ⇧ while clicking onto the arrange area with the pencil tool to the right of the track name in the Track List.
- This will launch a file browser, allowing you to select the desired audio file. Click once on the audio file you wish to use and press the **Open** button. The file will be inserted at the selected location.
  It may be worthwhile setting up a cycle region in the Arrange window, allowing you to continually cycle the audio part. This will make experimentation easier. Tip
- Open the Mixer window, or the Environments Audio layer, if you're running Logic Gold or Platinum. You may use the Windows menu or a Key Command (: ⌘ M/PC: ctrl M), or can simply double-click on the Audio Instrument track name to launch the Audio layer of the Environment window.
- Click-hold on any of the plug-in slots of the track's Audio channel and a hierarchical menu will open.
- Browse to the **EVOC 20 TO** *or* **EVOC 20 FB** entries, found in the **Filter** group of the plug-in list. Once the desired effect name is selected (highlighted), release the mouse button.
- This will automatically launch the plug-in window provided that the **Audio > Audio Preferences … > Display > Open plug-in window on insertion** parameter is active. We recommend that this parameter be activated. By default, it is **ON**. Tip
- You can manually open the EVOC20 plug-in window (at any time) by double-clicking on the blue EVOC20 plug-in Insert field of the channel.
- Once the EVOC20's graphical interface is launched, press the play button on the Transport Bar, or use the *Play* Key Command (0 on the numeric keypad).
- And now … as the file is playing back, do a little experimentation with the knobs, sliders and other controls. Have fun,

and feel free to insert further effect plug-ins on the channel or busses to further enhance and manipulate the sound.

- Changes made to the parameters of the EVOC 20 can be saved and later recalled. Please refer to the *Saving and Selecting Settings* section, from page 24 onwards, for further information.

It should be noted that the EVOC 20 TO uses a Side Chain, allowing the use of another track as the analysis and/or synthesis signal. In the gray area at the top of the plug-in window, click-hold on the **Side Chain** flip menu, and select the desired Audio track. In the Mixer, adjust the volume levels of the EVOC 20 and the audio track used for the *Side Chain* to taste.

4 The EVOC20—Basics

What is a Vocoder?

The word "Vocoder" is an abbreviation for VOice enCODER. As with many technologies in this otherwise beautiful world, it is a child of war. The Vocoder was initially developed for secure speech transmission over telephone lines which couldn't be "tapped". To achieve this, the speech signal was analyzed and only the "cryptic" results of the analysis were transmitted over telephone lines. On the receiving side, these results were used to synthetically rebuild the original voice signal.

Fortunately, Vocoders are used nowadays for altogether more peaceful purposes—namely for music. A Vocoder analyses and transfers the sonic character of the audio signal arriving at its analysis input to the audio signal present at its synthesis input. The result of this process is heard at the output of the Vocoder.

It should be noted that *any* audio signal can be analyzed: A Vocoder is not limited to speech signals.

How does a Vocoder work?

The speech *analyzer* and *synthesizer* referred to above are actually two *filter banks* of "band pass" filters. Band Pass filters allow a frequency band (a slice) in the overall frequency spectrum to pass through unchanged, and "cut" the frequencies which fall outside of the band's range.

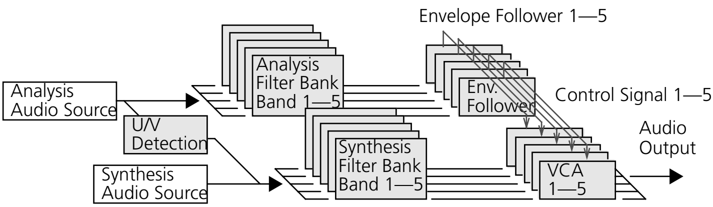
In the EVOC20, these filter banks are named the *Analysis* and *Synthesis* sections. These filter banks have a matching number of corresponding "bands"—i. e. if the *Analysis* filter bank has five "bands" (1, 2, 3, 4 and 5), there will be a corresponding set of five "bands" in the *Synthesis* filter bank. "Band" 1 in the *Analysis* bank is matched to "band 1" in the *Synthesis* bank, "band 2" to "band 2", etc.

In the EVOC 20, the audio signal arriving at the analysis input passes through the *Analysis* filter bank where it is divided into up to 20 bands.

An *envelope follower* is coupled to each filter band. The *envelope follower* of each band “tracks” (follows) any volume changes in the portion of the audio source allowed to “pass” by the associated band pass filter. In this way, the *envelope follower* of each band generates dynamic *control signal*s.

These *control signal*s are then sent to the *Synthesis* filter bank where they control the levels of the corresponding *Synthesis* filter bands. This is done via *VCAs—Voltage Controlled Amplifiers*. This allows the volume changes of the bands—and thus the changes of the original sound—in the *Analysis* filter bank to be imposed on the matching bands in the *Synthesis* filter bank.

The more bands a Vocoder offers, the more precisely the original sound’s character will be re-modeled.



Analyzing Speech Signals

The principles you’ve been introduced to thus far are insufficient for the transmission of speech signals. The reason is that human speech consists of a series of *vowels* (voiced, tonal sounds) and *consonants* (unvoiced, noisy sounds). The main distinction between vowels and consonants is that vowels are produced by an oscillation of the vocal cords, while consonants are produced by blocking and restricting the air flow with lips, tongue, palate, throat and larynx.

Should speech containing consonants and vowels be used as a Vocoder’s analysis signal, but the synthesis engine doesn’t

differentiate between voiced and unvoiced sounds, the result will sound rather “toothless”. To avoid this, the synthesis section of the Vocoder must produce different sounds for the *voiced* and *unvoiced* parts of the signal.

An example: In the word “synthesis”, the *unvoiced* portions of the word would be as follows:

“ess” as in *“S” ynthe “S” i “S”* and
“th” as in syn “TH” esis

In the EVOC 20 TO and PS plug-ins, there is an Unvoiced/ Voiced detector. This unit detects the unvoiced portions of the sound in the analysis signal and then substitutes the corresponding portions in the synthesis signal with *Noise*, a mixture of *Noise + Synth* or with the original signal (*Blend*). If the U/V Detector detects voiced parts, it passes this information to the *Synthesis* section, which uses the normal synthesis signal for these portions. Control over unvoiced/voiced sound detection, type and level is found in the *U/V Detection* section of these two EVOC 20 plug-ins.

Tip

It should be noted that an “s” sound contains a lot of high frequency content, whereas “p” or “b” sounds contain a lot of low frequency energy. Due to the way human beings hear, the intelligibility of speech is highly dependent on the presence of high frequency content. To aid in keeping speech clear, it may be worthwhile using equalization to boost or cut particular frequencies in analysis signals before processing them with the EVOC 20 PS or TO. Please see the *Tips for Better Speech Intelligibility* chapter, from page 63 onwards, for further information.

How does a Filter Bank work?

If you removed all circuits responsible for transferring the sonic characteristics from the analysis to the synthesis signal from a Vocoder, and dispensed with the detection of voiced or unvoiced signals, you’d be left with two filter banks—the analysis and synthesis filters.

To use these musically, you would need to ensure that you could control the output level of each band pass filter. With this level of control, you can apply unique and dramatic changes to the frequency spectrum.

5 Overview and Integration

Multiple EVOC 20 instances can be opened simultaneously. Each EVOC 20 PS instance requires its own Audio Instrument channel. The number of instances which can be opened simultaneously is entirely dependent on your Logic version, and processing resources available. The type and speed of the CPU used, and any other tasks—such as additional effects processing, Audio Instrument use or Audio track playback—performed simultaneously will affect performance.

All parameters of each EVOC 20 instance, and all associated mixer parameters can be fully automated. The EVOC 20 responds to automation data, allowing you to easily edit or create automation data in any of Logic's suitable editor window(s). Please refer to your Logic reference manual for further information.

To save precious processing resources, you can record the EVOC 20 signals (including automation) to disk at any time via Logic's **Bounce** function. The resulting audio files generated by this process can then be used on Audio Tracks within your arrangement. This type of functionality may prove of use when a song requires more processing power than your CPU is capable of delivering.

The Plug-in Window

Hands-on operation of all EVOC 20 effects is performed in the plug-in window. Double-click on the EVOC 20 insert panel on the Audio Object channel strip to access the plug-in window. Each instance of the individual EVOC 20 effects offers a discrete plug-in window, allowing each instance to have unique settings.

The parameters described in the following chapters are easier to manipulate from within the **Editor** view of the plug-in window. If you can see multiple horizontal sliders on a blue background, please switch from the **Controls** view to the **Editor**

view, using the flip menu found in the upper portion of the plug-in window.

Common Plug-in Window Parameters

Common to all of the EVOC 20 and other Logic plug-ins is the gray area shown at the top of the plug-in window. The EVOC 20 TO and EVOC 20 PS feature a **Side Chain** flip menu.

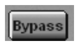
Link

If the Link button is switched off, you can open several plug-in windows simultaneously.

If the Link button is switched on (default), a single plug-in window will be used to display all opened plug-ins.

Each time you launch a new plug-in, the window will update to reflect the new selection. You can easily and conveniently switch between active plug-ins without relaunching the plug-in window. Please see the *Switching the Contents of the Plug-in Window* section, from page 25 onwards.

Bypass



The Bypass button bypasses the effect plug-in, e.g. for testing purposes.

Saving and Selecting Settings

The EVOC 20 saves any parameter changes you make in the plug-in window as a *Setting*. Every plug-in available for use in your version of Logic allows the storage and recall of these parameter changes.

Settings for the EVOC 20 processors are saved and recalled from the corresponding *EVOC 20 TO*, *EVOC 20 PS* and *EVOC 20 FB sub-folders* within the *Plug-In Settings* folder.

Save Setting

- To save a Setting after making parameter changes, click on the Settings flip menu and select **Save Setting** or **Save As... Setting**.
- In the ensuing file save dialog, type in the desired name, then click the save or OK button. This will automatically save the Setting in the appropriate folder for the selected EVOC20 effect.

It is strongly recommended that you do not attempt to change the folder structure *Plug-in Setting > EVOC20*. Inside the corresponding EVOC20 folder you are, however, free to sort your settings in sub folders. This folder structure is reflected in a hierarchical menu each time you load a plug-in setting.

As with all other plug-ins in Logic, the EVOC20 parameter changes will be saved with the song as well. When the song is re-opened, the stored parameter changes will be recalled.

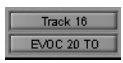
Load Setting

- To load an EVOC20 Setting, click on the Settings flip menu found above the EVOC20 panel.
- Scroll to the **Load Setting** option. In the ensuing dialog window, browse to and select the name of the Setting that you wish to load and click once. The Setting will then load.

Note that if no Setting preset is loaded, a default set of parameters will be used for each of the EVOC20 plug-ins.

Switching the Contents of the Plug-in Window

You can reassign any open plug-in window in two different ways using the two flip menus to the right of the *Settings* flip menu:



- Using the upper flip menu (*Track 16* in the diagram), you can switch the editor window between all channels. If you have inserted the EVOC20 FB on tracks 1 and 6, for example, you can switch between these channels and adjust their effect parameters.

- In the lower flip menu you can switch between the plug-ins slots of the selected channel, i. e. if a particular channel uses a Chorus and an EVOC 20 plug-in, you can switch the window between them.

Changing the Plug-In View

The **Editor** button allows you to switch between the graphical look and the **Controls** look (horizontal faders on a blue background).

Side Chain

The **Side Chain** flip menu allows the selection of an audio track. This facilitates the routing of an audio signal from another audio channel into the plug-in via a "side chain". In the EVOC 20, this facility enables the use of one audio track to be used as the *Analysis* source signal for the processing of another audio track (EVOC 20 TO). This can lead to many interesting and creative options and results.

ℹ Note that the **Side Chain** flip menu is only found on the EVOC 20 TO and PS plug-ins.

Hints for Changing Parameters

- You can reset any parameter to its default value by ⌥-click (Mac) or ctrl-click (Windows).
- If you hold ⇧ before clicking and moving a control, its value can be fine-tuned.
- To control a slider, you can click in a free part of the slider way and move the mouse, too. This works also, if it is a double slider.

Automation

As with every Logic plug-in, the EVOC 20 effects can be fully automated. Please refer to your Logic documentation for further information.

6 EVOC20 PS Parameters

The EVOC20 PS is a sophisticated vocoder equipped with a polyphonic synthesis engine, capable of receiving MIDI note input. This allows the EVOC20 PS to be “played”, resulting in “classic” vocoder choir sounds, for example. Single notes and chords played with the polyphonic EVOC20 PS will “sing” with the articulation of the *Analysis* audio source.

In this process, the sonic characteristics and changes of the audio signal arriving at the analysis input are imposed on the output signal of the integrated synthesizer (the *Synthesis* section).

As outlined earlier, the EVOC20 PS can only be used in the *top* insert slot of an Audio Instrument channel. Please refer to the *Using the EVOC20 PS* section, from page 15 onwards, for step-by-step instructions on the insertion of this plug-in.

The EVOC20 PS interface is divided into six main sections. These are the *Synthesis*, *Sidechain Analysis*, *Formant Filter*, *Modulation*, *U/V Detection* and *Output* areas.

Synthesis Parameters

The EVOC 20 PS is equipped with a polyphonic synthesis engine. It is the only EVOC 20 plug-in capable of accepting MIDI note input. The parameters of the *Synthesis* section are described below.

Mode Switches

These switches determine the number of voices used by the EVOC 20:

- When **Poly** is selected, the maximum number of voices is set via the numeric field alongside the **Poly** button. To change the value, click and hold with your mouse, and drag up or down to increase/decrease polyphony.

It should be noted that increasing the number of voices also increases processor overhead.

- When **Mono** or **Legato** is selected, the EVOC 20 is monophonic, and uses only one voice.
  - In **Legato** mode, **Glide** (see page 32) is only active on tied notes. Envelopes are not retriggered when tied notes are played (single trigger).
  - In **Mono** mode, **Glide** is always active and the envelopes are retriggered by every note played (multi trigger).
- The **Unison** button enables/disables unison mode. In this mode, each EVOC 20 PS voice is doubled, which will cut polyphony in half (to a maximum of 8 voices) as indicated by the numeric **Voices** field. The doubled voices are detuned by the amount defined with the **Analog** parameter. (Also see the *Analog Tuning* section, from page 32 onwards.)
- In *Unison-Mono* mode (both the **Unison** and **Mono** or **Legato** buttons are active), up to 16 voices can be stacked and played monophonically. In this mode, the **Voices** field displays the number of stacked voices that sound at the same time.

Stacking voices in *Unison-Mono* mode will increase the EVOC 20's output volume. To avoid overloading the Audio Instrument channel, adjust the EVOC 20 *Level* slider accordingly.

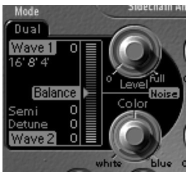
Oscillator Section

The EVOC 20 PS is equipped with a two oscillator digital synthesizer which features a number of waveforms, and FM (Frequency Modulation). Further to these sound-generators in the *Synthesis* section is an independent *Noise* generator.

There are two oscillator modes.

- **Dual**: Where two oscillators make use of single-cycle digital waveforms to provide the *Synthesis* sound source(s).
- **FM**: A two operator FM engine, with Oscillator 1 as a sine wave carrier, and Oscillator 2 as the modulator. Oscillator 2 can use any of the single-cycle digital waveforms.

You can switch between **Dual** and **FM** modes by clicking on the **Dual** or **FM** label(s) to the top left of the section shown in the diagrams.



As you can see, there are some subtle differences between the two modes. We will look at the common parameters first, and will then look at the mode-specific options.

Wave 1 Parameters

The "footages" below the **Wave 1** label in both modes harks back to the days of pipe organs. The longer the pipe, the deeper the tone. This also applies to *Wave 1*. Simply click on the **16**, **8** or **4** foot value to select the range in which Wave (oscillator) 1 functions. Your selection will be illuminated.

The numerical value beside the **Wave 1** label (shown as **41** in the diagrams) indicates the currently selected waveform type. The EVOC 20 PS features 50 waveforms with different sonic characteristics. To switch between them, simply click-hold on the numerical field and drag up or down. When the desired waveform number is visible, release the mouse button.

It should be noted that when in FM mode, the waveform of **Wave 1** is a fixed sine wave. The waveform parameter of **Wave 1** does not have an effect in this mode.

Wave 2 Parameters

The numerical value beside the **Wave 2** label (shown as "41" in the diagrams) indicates the currently selected waveform type. The EVOC 20 PS features 50 single-cycle digital waveforms with different sonic characteristics. To switch between them, simply click-hold on the numerical field and drag your mouse up or down. When the desired waveform number is visible, release the mouse button.

Noise Parameters

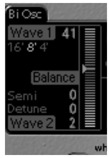
The *Noise* generator provides a further sound source which can be used *in addition to* the two oscillators (*Wave 1* and *Wave 2*).

The **Level** knob controls the amount of noise added to the signals of the two oscillators, and the **Color** knob controls the timbre of the noise signal. When the **Color** knob is turned full-left, the *Noise* generator creates a pure *white noise*. When turned full-right, it generates *blue noise* (high-passed noise). White noise always has been used to create wind and rain sound effects. It has the same energy in each frequency interval. Blue noise sounds brighter, since its bass portion is suppressed by a high pass filter.

It is important to note that the *Noise* generator in the *Oscillator* section is independent of the *Noise* generator in the *U/V Detection* section. For further information on voiced and unvoiced signals, please refer to the *Analyzing Speech Signals* section, from page 20 onwards.

Turn **Color** full-right and **Level** just a tiny bit up to achieve a more lively and "fresh" synthesis signal.

Dual Mode Parameters

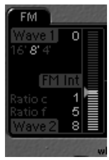


The parameters specific to the **Dual** mode are found in the *Wave 2* section, and the **Balance** slider to the right.

- The **Semi** parameter adjusts the tuning of the second oscillator (Wave 2) in semitone steps. Adjustment is made by using the mouse as a slider directly on the numerical field. Its Range: **±24**, or up/down two octaves, respectively.
- The **Detune** parameter fine tunes *Wave 1* and *Wave 2* in cents. 100 cents equals a semitone step. Doing so will detune *Wave 1* in conjuction with *Wave 2* around the tuning zero point. The range is **±50** cents, or "up/down half a semitone". Adjustment is made by using the mouse as a slider directly on the numerical field.
- The **Balance** slider allows you to blend the two oscillators (*Wave 1* and *Wave 2*).

FM Mode Parameters

The parameters specific to the FM mode are found in the *Wave 2* section, and the **FM Int** slider to the right.



- The **Ratio c(oarse)** parameter adjusts the coarse frequency ratio of the second oscillator in relation to the first oscillator. Adjustment is made by using the mouse as a slider directly on the numerical field. Range: **0—32**.
- The **Ratio f(ine)** parameter adjusts the fine frequency ratio of the second oscillator in relation to the first oscillator. The range is **0—99**. Adjustment is made by using the mouse as a slider directly on the numerical field
- The **FM Int** slider determines the intensity of *Wave 1's* sine wave modulation by *Wave 2*. Higher *FM Int.* settings will result in a more complex waveform with more overtones.

When combined, the **Ratio** and **FM Int** parameters form the resulting complex FM waveform and thus define their harmonic content.

Analog Tuning

The **Analog** tuning parameters simulate the instability of analog circuitry found in vintage vocoders. **Analog** alters the pitch of each note randomly. This behavior is much like that of polyphonic analog synthesizers. The **Analog** knob controls the intensity of the random detuning.

Tuning

The range of detuning is defined in the **Tune** window. Adjustments are made by using the mouse as a slider. The range is from **425** to **455Hz**.

Glide

The effect of this knob depends on the setting made in the **Bend Range** window. **Glide** determines the time it takes for the pitch to slide from one note to another (portamento). The range is **+5000 ms**.

Bend Range

Bend Range determines the pitch range in semitones for pitch bend modulation. The range is ±12 semitones.

Cutoff

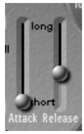
The cutoff frequency of the lowpass filter. As you turn this knob to the left, an increasing number of high frequencies are filtered from the signal.

Resonance

Turning up **Resonance** leads to an emphasis of the frequency area surrounding the frequency defined by the **Cutoff** parameter.

i The filter is used for rough signal shaping, before the signal is articulated by the vocoding circuits. Hint: Set Cutoff as high as possible and dial in a little bit of Resonance to get a nice, brilliant high-end.

Envelope



This is an *Attack/Release* envelope generator used for level control over the oscillator section.

- The **Attack** parameter determines the amount of time that it takes for the *Oscillators* of the *Synthesis* section to reach their maximum level.
- The **Release** parameter determines the amount of time that it takes for the *Oscillators* of the *Synthesis* section to reach their minimum level.

Sidechain Analysis In Parameters

Attack

The **Attack** parameter determines, how fast each *envelope follower* coupled to each *Analysis* filter band reacts to rising signals. Longer **attack** times result in a slower tracking response to transients of the *Analysis* input signal.

A long **attack** time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. Set **Attack** as low as possible to get precise articulation.

Release

The **Release** parameter determines, how fast each *envelope follower* coupled to each *Analysis* filter band reacts to falling signals. Longer **Release** times make transients of the *Analysis* input signal sound longer at the vocoder's output.

A long **release** time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. But **Release** times that are too short result in rough, grainy

vocoder sounds. **Release** values of around **8** to **10** ms have proven to be useful starting points.

Freeze

The **Freeze** button holds the current *Analysis* sound spectrum infinitely.

The "frozen" *Analysis* signal can capture a particular characteristic of the source signal which is then imposed as a complex sustained filter shape on the *Synthesis* section.

Using a spoken word pattern as a source, for example, the **Freeze** parameter could capture the attack or tail phase of an individual word within the pattern—e.g. the vowel "a".

With **Freeze** engaged, the *Analysis* filter bank ignores the input source until it is disengaged.

Another use of the **Freeze** parameter (which can be automated) could be to compensate for people's "inability" to sustain sung notes for a long period without taking a breath. If the *Synthesis* signal needs to be sustained, when the *Analysis* source signal (a vocal part) isn't, **Freeze** can be used to "lock" the current formant levels (of a sung note) even during gaps in the vocal part—i. e. between words in a vocal phrase.

When the **Freeze** parameter is used, the **Attack** and **Release** parameters have no effect.

Bands

The **Bands** window determines the number of frequency bands used by the EVOC 20 PS. It ranges from **5** to **20**. Adjustments are made by using the mouse as a slider.

The greater the number of bands, the more precisely the sound can be reshaped. As the number of bands is reduced, the source signal's frequency range is divided up into fewer bands—and the resulting sound will be formed with less precision by the *Synthesis* engine.

Tip

It should be noted that increasing the number of **Bands** also increases the processor overhead. You may find that a good compromise between sonic precision—allowing incoming signals (speech and vocals, in particular) to remain intelligible—and resource usage, is around **10** to **15** bands.

Formant Filter Parameters

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the *Analysis* section and the lower half to the *Synthesis* section. Changes made to the **High/Low** frequency parameters, the **Bands** parameter or the **Formant Stretch** and **Shift** parameters will result in visual changes to the Formant Filter window. This provides you with invaluable feedback on what is happening to the signal as it is routed through the two Formant Filter banks.

High/Low Frequency

The blue bar shown just beneath the **emagic** logo is a multi-part control which is used to determine the lowest and highest frequencies allowed to “pass” by the filter section. The length of the blue bar represents the frequency range for the analysis and synthesis. Frequencies of any audio input which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the **High/Low Frequency** values.

- To adjust the low frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is **75—750 Hz**.
- To adjust the high frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is **800—8000 Hz**.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the **High/Low Frequency** values directly by using your mouse as a slider on the numerical entries—**270** and **7100Hz** in the diagram.

Lowest/Highest

These parameters can be found in the two small “windows” on either side of the Formant Filter window. These switches determine whether the lowest and highest filter bands are band pass filters (just like all the bands between them), or whether they act as low pass/high pass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the **Band Pass** setting, the frequencies below/above the lowest/highest bands are ignored on analysis and synthesis.
- In the **High Pass** (or **Low Pass**) setting, all frequencies below the lowest (or above the highest) bands will also be considered for analysis and synthesis.

Formant Stretch

Alters the width and distribution of all bands in the *Synthesis* filter bank, extending or narrowing the frequency range defined by the blue bar (**Low/High Frequency** parameters) for the *Synthesis* filter bank.

With **Formant Stretch** set to **0**, the width and distribution of the bands in the *Synthesis* filter bank equal the width of the bands in the *Analysis* filter bank. Low values narrow the width of each band, while high values widen it. The control range is from **0.5** to **2** (as a ratio of the overall bandwidth).

You can jump directly to the value **1** by clicking on its number.

Formant Shift

Moves the position of all bands in the *Synthesis* filter bank up and down. With **Formant Shift** set to **0**, the position of the bands in the *Synthesis* filter bank equal the position of the bands in the *Analysis* filter bank. Positive values will move the bands up in frequency, while negative values will move them down in respect to the *Analysis* filter bank.

You can jump directly to the values **–0.5**, **–1**, **0**, **+0.5** and **+1** by clicking on their numbers.

When combined, **Formant Stretch** and **Formant Shift** alter the formant structure of the resulting vocoder sound and can result in some interesting timbre changes. For example, using speech signals and tuning **Formant Shift** up results in Mickey Mouse effects.

Formant Stretch and **Formant Shift** are also useful if the frequency spectrum of the *Synthesis* signal does not complement the frequency spectrum of the *Analysis* signal. You could create a synthesis signal in the high frequency range from an analysis signal which mainly modulates the sound in a lower frequency range, for example.

Resonance

Resonance is responsible for the basic sonic character of the vocoder: low settings give it a soft character, high settings will lead to a more snarrling, sharp character. Increasing the value for **Resonance** emphasizes the middle frequency of each frequency band.

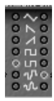
The use of either, or both, of the **Formant Stretch** and **Formant Shift** parameters can result in the generation of unusual resonant frequencies when high **Resonance** settings are used.

Modulation Parameters

The Modulation (*LFO*) area offers two LFOs to control the **Formant Shift** and **Pitch** parameters of the EVOC 20 PS. The LFOs can run free or synchronized to the song's tempo.

- *Pitch LFO*, on the left-hand side, controls *Pitch* modulation (Vibrato) of the built-in synthesizer's oscillators. It is "hard wired" to accept data from the Mod Wheel of your MIDI keyboard or from corresponding MIDI data to control modulation intensity.
- *Shift LFO* controls the **Shift** parameter of the *Synthesis* filter bank to produce dynamic phasing-like effects.

Wave



These two switches allow the selection of the waveform type used by *Pitch LFO* and *Shift LFO*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H, "random music"), and a smoothed random waveform is available for each LFO.

Intensity/Int via Whl

The **Intensity** slider controls the amount of **Formant Shift** modulation by *Shift LFO*.

The **Int via Whl** slider for the *Pitch LFO* features a multi-part slider. The intensity of LFO pitch modulation can be controlled by the Modulation wheel on an attached MIDI keyboard. The upper half of the slider determines the intensity when the Modulation wheel is set to its maximum value, and the lower half when set to its minimum value. By clicking and dragging in the area between the two slider segments, you can simultaneously move both.

Rate Knobs

These knobs determine the speed of the modulation. Values to the left of the center positions are synchronized with the sequencer's tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous and displayed in Hertz (cycles per second).

The ability to use synchronous bar values could be used to perform a formant shift every four bars on a one bar percussion part, which is being cycled. Alternately, you could perform the same formant shift on every eighth note triplet within the same part. Either method can generate interesting results, and can lead to new ideas, or a new lease of life on old audio material.

Unvoiced/Voiced (U/V) Detection

Please refer to the *Analyzing Speech Signals* section, from page 20 onwards, for an explanation of the *U/V Detection* principle.

Speech intelligibility is highly dependent on high frequency content, as human hearing is reliant on them to determine syllables within words. Bear this fact in mind when using the EVOC 20, and take care with filter frequency settings in the ***Synthesis*** and ***Formant Filter*** sections.

To aid intelligibility, it may be worthwhile using equalization to boost particular frequencies in the mid to high frequency range, before processing the signal with the EVOC 20 PS. Please see the *Tips for Better Speech Intelligibility* chapter, from page 63 onwards, for further information.

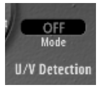
Sensitivity

Determines how responsive the U/V detection is. By turning this knob to the right, more of the individual *unvoiced* (consonants) portions of the input signal are recognized.

When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V source—determined by the **Mode** parameter—being used on the majority of the input signal, including *voiced*

signals. Sonically, this results in a sound that resembles a radio signal which is "breaking up" and contains a lot of "static" or noise.

Mode (Flip Menu)



Here, you can select the sound source(s) which can be used to replace the *unvoiced* content of the input signal. Possible settings are **Off**, **Noise**, **Noise + Synth** or **Blend**.

- **Noise**—uses noise alone for the unvoiced portions of the sound.
- **Noise + Synth**—uses noise and the synthesizer for the unvoiced portions of the sound,
- **Blend**—uses the analysis signal after it has passed through a high-pass filter, for the unvoiced portions of the sound. This filtered analysis signal is then always mixed with the EVOC 20 output signal. The **Sensitivity** parameter has no effect when this setting is used.

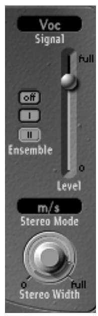
Level

The **Level** knob controls the amount/volume of the signal (**Noise**, **Noise** + **Synth or Blend**) used to replace the *unvoiced* content of the input signal.

Care should be taken with this control, particularly when a high *Sensitivity* value is used, to avoid internally overloading the EVOC 20.

Output Parameters

Signal



This flip menu offers the choice of **Voc(oder)**, **Syn(thesis)** and **Ana(lysis)**. Selection of one of these allows you to determine which signal you wish to be sent to the EVOC 20's main outputs. To hear the vocoder effect, the **Signal** parameter should be set to **Voc**. The other two settings are useful for monitoring purposes.

Ensemble

The three **Ensemble** switches switch the ensemble effect(s). **Ensemble I** is a special chorus effect. **Ensemble II** is a variation, creating a fuller and richer sound by using a more complex modulation routine. Simply click on the desired button to activate the preferred effect.

Level

The **Level** slider controls the amount/volume of the EVOC20's output signal.

Stereo Width

Stereo Width distributes the output signals of the *Synthesis* section's filter bands in the stereo field.

- In the **left** position, the output of all bands are centered.
- In the **center** position, the output of all bands ascends from left to right.
- In the **right** position, the bands are output—alternately—on the left and the right channel.

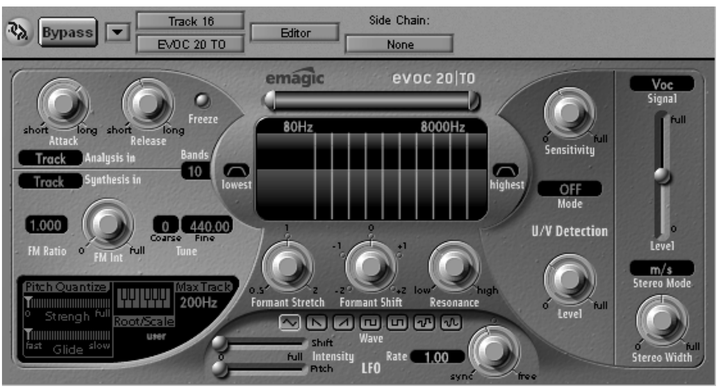
7 EVOC20 TO Parameters

The EVOC20 TO is a Vocoder equipped with a monophonic pitch *tracking oscillator*, hence the "TO" in its name. Non-technically, this allows the EVOC20 TO to follow (track) the pitch of a monophonic incoming audio signal. If the audio signal is a vocal melody, for example, the individual pitches of the sung notes will be tracked and mirrored by the *Synthesis* engine.

Please note: For good pitch tracking it is absolutely essential that the signal is monophonic (one pitch at a time) and as dry as possible. Absolutely avoid background noises. For example, using a voice already processed with slight reverb will give pretty strange results. The results will be even stranger, when signals with no audible pitch are used—like drum loops. However, the resulting artefacts might be exactly what you are after in some situations.

But the EVOC20 TO is not limited to pitch-tracking effects. It can vocode a signal by itself, very useful for unusual filter effects. Try this with different **Resonance**, **Formant Shift** and **Formant Stretch** settings.

And as both analysis and synthesis input signals are freely selectable, you can even vocode, for example, an orchestra with train noises.



The EVOC20 TO can be used in the insert slots of Audio, Audio Input, Bus, Master and Audio Instrument channels. Please refer to the *Using the EVOC20 TO and EVOC20 FB* section, from page 17 onwards, for step-by-step instructions on the insertion of this plug-in.

The EVOC20 TO interface is divided into five main sections. From left to right, these are the *Analysis/Synthesis*, *Formant/Filter*, *Modulation*, *Unvoiced/Voiced (U/V) Detection* and *Output* areas.

Analysis In Parameters

Attack

The **Attack** parameter determines, how fast each *envelope follower* coupled to each *Analysis* filter band reacts to rising signals. Longer **attack** times result in a slower tracking response to transients of the *Analysis* input signal.

A long **attack** time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. Set **Attack** as low as possible to get precise articulation.

Release

The **Release** parameter determines, how fast each *envelope follower* coupled to each *Analysis* filter band reacts to falling signals. Longer **Release** times make transients of the *Analysis* input signal sound longer at the vocoder's output.

A long **release** time on percussive input signals (a spoken word or hi-hat part, for example) will translate into a less articulate vocoder effect. But **Release** times that are too short result in rough, grainy vocoder sounds. **Release** values of around **8** to **10** ms have proven to be useful starting points.

Freeze

The **Freeze** button holds the current analysis sound spectrum indefinitely.

The "frozen" *Analysis* signal can capture a particular characteristic of the source signal, which is then imposed as a complex sustained filter shape on the *Synthesis* section.

Using a spoken word pattern as a source, for example, the **Freeze** parameter could capture the attack or tail phase of an individual word within the pattern—e. g. the vowel "a".

With **Freeze** engaged, the *Analysis* filter bank ignores the input source until it is disengaged.

Another use of the **Freeze** parameter (which can be automated) could be to compensate for people's "inability" to sustain sung notes for a long period without taking a breath. If the *Synthesis* signal needs to be sustained, when the *Analysis* source signal (a vocal part) isn't, *Freeze* can be used to "lock" the current formant levels (of a sung note) even during gaps in the vocal part—i. e. between words in a vocal phrase.

When the **Freeze** parameter is used, the **Attack** and **Release** parameters have no effect.

Analysis In (Flip Menu)

This flip menu determines the *Analysis* signal source—**Track** or **Side Chain**. To switch between them, use the mouse as a slider and drag vertically.

- **Track**—allows you to use the audio track, into which the EVOC 20 is inserted, as the analysis source signal.
- **Side Chain**—allows you to use another audio track as the analysis source signal. The selection of the actual Side Chain source track is achieved by click-holding on the *Side Chain* flip menu in the gray area at the top of the plug-in window.

Tip

If no Side Chain track is assigned in the Side Chain flip menu, then the track signal will be used.

Synthesis In Parameter

Synthesis In (Flip Menu)

This flip menu determines the *Synthesis* signal source—**Osc(illator)**, **Track** or **Side Chain**. To switch between them, use the mouse as a slider and drag vertically.

- **Oscillator**—allows you to use the built-in monophonic tracking oscillator. The oscillator tracks the pitch of the *Analysis* input signal. Selection of the **Oscillator** activates the other parameters in the *Synthesis* section. If **Osc** is not selected, the **FM Ratio**, **FM Int** and other parameters in this area have no effect.
- **Track**—allows you to use the audio track, into which the EVOC 20 TO is inserted, as the *Synthesis* source signal.
- **Side Chain**—allows you to use another audio track as the source material for the *Synthesis* section. Selection of the Side Chain track is achieved by click-holding on the *Side Chain* flip menu in the gray area at the top of the plug-in window.

Tip

If no Side Chain track is assigned in the Side Chain flip menu, then the track signal will be used.

Bands

The **Bands** window determines the number of frequency bands used by the EVOC 20 TO. It ranges from **5** to **20**. Adjustments are made by using the mouse as a slider. The greater the number of bands, the more precisely the sound can be reshaped.

Tip

Increasing the number of bands also increases the processor overhead.

The Tone Generator of the Tracking Oscillator

Depending on the position of the **FM Int** control, the tracking oscillators delivers either a sawtooth wave or the signal of an FM tone generator.

ℹ The FM tone generator consists of two oscillators, each generating a sine wave. The frequency of Oscillator 1 is linearly modulated by Oscillator 2. This deforms the sine wave of oscillator 1 to a waveform with rich harmonic structure. Its harmonic structure depends on the modulation intensity and the frequency ratio of both oscillators.

Tune

Coarse Tune offsets the pitch of the oscillator in semitones by up to ±2 octaves.

Fine Tune: The default value is concert pitch A = 440 Hz. The range is from **425.00** to **455.00 Hz**.

FM Int(ensity)

This knob selects the basic waveform and controls the intensity of the FM modulation.

- If set to **0**, the FM tone generator is disabled and a sawtooth wave is generated instead.
- If set to values higher than **0**, the FM tone generator is activated. Higher values result in more complex and brighter sound.

FM Ratio

The **FM Ratio** (value range **0.5** to **3.5**) knob defines the ratio between the Carrier and Modulator frequencies—i. e. the frequencies of Oscillators 1 and 2. This setting defines the basic character of the sound.

With even-numbered values or their multiples, harmonic sounds are produced. With odd-numbered values or their

multiples, inharmonic sounds are produced, which we perceive as being “metallic” sounding.

- An **FM Ratio** of 1.000 produces results resembling a sawtooth waveform.
- An **FM Ratio** of 2.000 produces results resembling a square wave with a pulse width of 50%.
- An **FM Ratio** of 3.000 produces results resembling a square wave with a pulse width of 33%.

This control is only relevant, if **FM Int** is not set to 0.

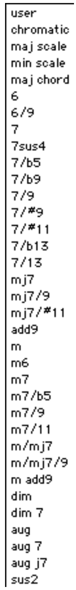
Pitch Quantize

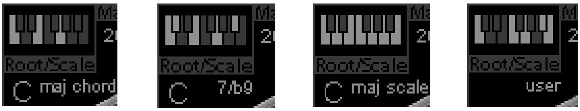
The **Pitch Quantize**, **Root/Scale** and **Max Track** controls, in conjunction with the piano keys of the onscreen keyboard, control the automatic pitch correction facility (Pitch Quantize) of the tracking oscillator. **Pitch Quantize**, in conjunction with the **Root Scale** and **Max Track** parameters, can be used to constrain the pitch of the tracking oscillator to a scale or chord. This allows subtle or savage pitch corrections, and can be used creatively on unpitched material with high harmonic content, such as cymbals and high-hats. To use pitch quantization, the **Strength** parameter must be set above a value of zero, and at least one of the onscreen keyboard keys needs to be activated.

- **Strength**—determines how pronounced the automatic pitch correction is.
- **Glide**—determines the amount of time the pitch correction takes, allowing “sliding” transitions to the quantized pitches.

Root/Scale

The **Root** and **Scale** parameters, in combination with the onscreen keyboard, define the pitch(es) that the tracking oscillator is quantized to.





- If you click-hold on the word **Scale** and drag vertically, you can select a scale or chord. See the listing of preset scales and chords shown alongside.
- **Root** selects the root key of the respective scale or chord. Note: The **Root** parameter is not displayed when **chromatic** or **user** is selected.
- Any combination of keys can be activated by clicking on the notes of the onscreen keyboard. Activated keys are illuminated. To disable any active notes on the keyboard, simply click on the note a second time. The **Scale** display will change to **user** as soon as any key is edited.
- The previously displayed scale or chord is used as the starting point when creating a user scale. This allows you to select a preset scale or chord, and then modify it by clicking on the notes of the onscreen keyboard.
- The last edit will be remembered. You can select a new preset scale or chord, and as long as you don't make any changes you can always jump back to the previously set **user** scale.

As with all Logic plug-ins, the **Root and Scale** parameters, and the keys of the onscreen keyboard can be automated.

Max Track

This parameter cuts the high frequencies of the analysis signal, making the pitch detection more robust. Should the pitch detection produce unstable results, reduce the **Max Track** parameter value to the lowest possible setting. Use the mouse as a slider to adjust the value.

Formant Filter

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the *Analysis* section and the lower half to the *Synthesis* section. Changes made to the **High/Low** frequency parameters, the **Bands** parameter or the **Formant Stretch** and **Shift** parameters will result in visual changes to the Formant Filter window. This provides you with invaluable feedback on what is happening to the signal as it is routed through the two Formant Filter banks.

High/Low Frequency

The blue bar shown just beneath the **emagic** logo is a multi-part control which is used to determine the lowest and highest frequencies allowed to "pass" by the filter section. The length of the blue bar represents the frequency range for the analysis and synthesis. Frequencies of any audio input which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the **High/Low Frequency** values.

- To adjust the low frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is **75—750 Hz**.
- To adjust the high frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is **800—8000 Hz**.

- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the High/Low frequency values directly by using your mouse as a slider on the numerical entries—**80** and **8000Hz** in the diagram.

Lowest/Highest

These parameters can be found in the two small “windows” on either side of the Formant Filter window. These switches determine whether the lowest and highest filter bands are band pass filters (just like all the bands between them), or whether they act as low pass/high pass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the **Band Pass** setting, the frequencies below/above the lowest/highest bands are ignored on analysis and synthesis.
- In the **High Pass** (or **Low Pass)** setting, all frequencies below the lowest (or above the highest) bands will also be considered for analysis and synthesis.

Formant Stretch

Alters the width and distribution of all bands in the *Synthesis* filter bank, extending or narrowing the frequency range defined by the blue bar (**Low/High Frequency** parameters) for the *Synthesis* filter bank.

With **Formant Stretch** set to **0**, the width and distribution of the bands in the *Synthesis* filter bank equal the width of the bands in the *Analysis* filter bank. Low values narrow the width of each band, while high values widen it. The control range is from **0.5** to **2** (as a ratio of the overall bandwidth).

ℹ You can jump directly to the value **1** by clicking on its number.

Formant Shift

Formant Shift moves the position of all bands in the *Synthesis* filter bank up and down. With **Formant Shift** set to **0**, the position of the bands in the *Synthesis* filter bank equal the position of the bands in the *Analysis* filter bank. Positive values will move the bands up in frequency, while negative values will move them down in respect to the *Analysis* filter bank.

You can jump directly to the values **–0.5**, **–1**, **0**, **+0.5** and **+1** by clicking on their numbers.

When combined, **Formant Stretch** and **Formant Shift** alter the formant structure of the resulting vocoder sound and can result in some interesting timbre changes. For example, using speech signals and tuning **Formant Shift** up results in Mickey Mouse effects.

Formant Stretch and **Formant Shift** are especially useful if the frequency spectrum of the *Synthesis* signal does not complement the frequency spectrum of the *Analysis* signal. You could create a synthesis signal in the high frequency range from an analysis signal which mainly modulates the sound in a lower frequency range, for example.

Resonance

Resonance is responsible for the basic sonic character of the vocoder: low settings give it a soft character, high settings will lead to a more snarrling, sharp character. Increasing the value for **Resonance** emphasizes the middle frequency of each frequency band.

The use of either, or both, of the **Formant Stretch** and **Formant Shift** parameters can result in the generation of unusual resonant frequencies when high **Resonance** settings are used.

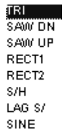
Modulation Parameters

The *LFO* can modulate …

- the *frequency* (“**Pitch**”) of the tracking oscillator (vibrato) or
- the **Shift** parameter of the *Synthesis* filter bank.

It allows synchronous/non-synchronous modulation in bar, beat (triplet) or free values.

Wave



The **Wave** switches allow the selection of a waveform type to be used by the *LFO*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar, good for trills), Square up from zero (unipolar, good for changing between two definable pitches), a random stepped waveform (S&H, “random music”), and a smoothed random waveform is available. Simply click on the appropriate button to select a waveform type.

Intensity

The **Intensity** sliders control the amount of **Formant Shift** and **Pitch** modulation (Vibrato) by the *LFO*.

Rate

These knob determines the speed of the modulation. Values to the left of the center positions are synchronized with the sequencer’s tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous and displayed in Hertz (cycles per second).

ℹ The ability to use synchronous bar values could be used to perform a formant shift every four bars on a one bar percussion part, which is being cycled. Alternately, you could perform the same formant shift on every eighth note triplet within the same part. Either method can

generate interesting results, and can lead to new ideas, or a new lease of life on old audio material.

Unvoiced/Voiced (U/V) Detection

Please refer to the *Analyzing Speech Signals* section, from page 20 onwards, for an explanation of the *U/V Detection* principle.

Speech intelligibility is highly dependent on high frequency content, as human hearing is reliant on them to determine syllables within words. Bear this fact in mind when using the EVOC 20, and take care with filter frequency settings in the ***Synthesis*** and ***Formant Filter*** sections.

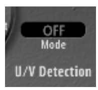
To aid intelligibility, it may be worthwhile using equalization to boost particular frequencies in the mid to high frequency range, before processing the signal with the EVOC 20 PS. Please see the *Tips for Better Speech Intelligibility* chapter, from page 63 onwards, for further information.

Sensitivity

Determines how responsive the U/V detection is. By turning this knob to the right, more of the individual *unvoiced* (consonants) portions of the input signal are recognized.

When high settings are used, the increased sensitivity to unvoiced signals can lead to the U/V source—determined by the **Mode** parameter—being used on the majority of the input signal, including *voiced* signals. Sonically, this results in a sound that resembles a radio signal which is "breaking up" and contains a lot of "static" or noise.

Mode (Flip Menu)



Here, you select the sound source(s) which can be used to replace the *unvoiced* content of the input signal. Possible settings are **Off**, **Noise**, **Noise + Synth** or **Blend**.

- **Noise**—uses noise alone for the unvoiced portions of the sound.

- **Noise + Synth**—uses noise and the synthesizer for the unvoiced portions of the sound,
- **Blend**—uses the analysis signal after it has passed through a high-pass filter, for the unvoiced portions of the sound. This filtered analysis signal is then always mixed with the EVOC 20 output signal. The **Sensitivity** parameter has no effect in this setting.

Level

Level controls the amount/volume of the signal (**Noise**, **Noise + Synth** or **Blend**) used to replace the *unvoiced* content of the input signal.

Care should be taken with this control, particularly when a high **Sensitivity** value is used, to avoid internally overloading the EVOC 20.

Output Parameters

Signal

This flip menu offers the choice of **Voc(oder)**, **Syn(thesis)** and **Ana(lysis)**. Selection of one of these allows you to determine which signal you wish to be sent to the EVOC 20's main outputs. To hear the vocoder effect, **Signal** must be set to **Voc**. The other two settings are useful for monitoring purposes.

Level

Level controls the amount/volume of the EVOC 20's output signal.

Stereo Mode

This flip menu determines the input/output mode of the *Synthesis* filter bank. Choices are: **m/s**—*mono* input to *stereo* output and **s/s**—*stereo* input to *stereo* output.

The **Stereo Mode** should be set to **m/s** if the signal going into the *Synthesis* filter bank is monophonic or **Synthesis In** is set to **Osc**.

Stereo/stereo *(**s/s**)* is the preferred setting for stereo *Synthesis* input signals. In this case, the stereo signal is processed by a separate filter bank for the left and right channels. When using the **m/s** Mode on stereo *Synthesis* input signals, the stereo signal is first summed to mono before it is passed on to the *Synthesis* filter bank.

Stereo Width

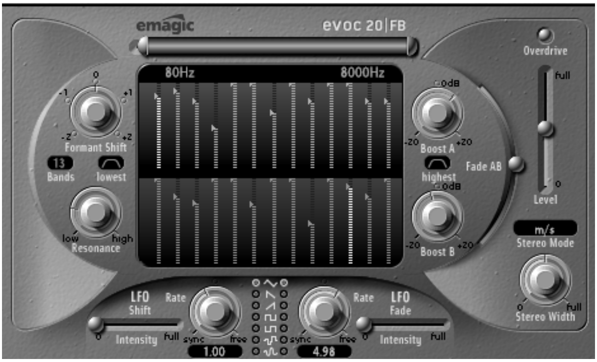
Stereo Width distributes the output signals of the *Synthesis* section's filter bands in the stereo field.

- In the left position, the output of all bands are centered.
- In the center position, the output of all bands ascends from left to right.
- In the right position, the bands are output—alternately—on the left and right channels.

The stereo/stereo mode (**s/s**) uses one filter bank per channel. The positioning of the frequency bands correspond to that described above, but the bands of each filter bank ascend in opposing directions, from left to right.

8 EVOC20 FB Parameters

The EVOC20 FB (*F*ilter *B*ank) is an extremely sophisticated combination of two filter banks—A and B. The input signal runs through both filter banks in parallel. You can manually or automatically crossfade between the output signals of both filter banks.



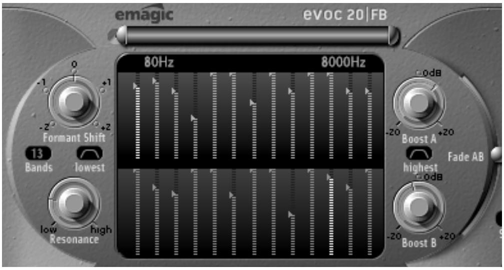
The EVOC20 FB can be used in the insert slots of Audio, Bus, Master and Audio Instrument channels. Please refer to the *Using the EVOC20 TO and EVOC20 FB* section, from page 17 onwards, for step-by-step instructions on the insertion of this plug-in.

The EVOC20 FB interface is divided into three main sections. These are the *Formant Filter*, *Modulation* and *Output* areas.

The Formant Filter Area

The Formant Filter Window

The Formant Filter window is divided into two sections by a horizontal line. The upper half applies to the “Filter Bank A”, and the lower half to the “Filter Bank B” .



The individual vertical bars in each bank of settings are “faders” which represent the level of a particular frequency band/formant. To adjust each “fader”, simply click-hold on the desired bar and drag up or down.

Complex “bar curves” are easily created by “painting” them in: Click and hold the mouse button next to a bar on the blue or green portion of the background, and drag left or right over the bars within the editing field. The length of the bars will be adjusted in accordance with the mouse movement. This method makes editing multiple frequency band levels quick and convenient.

Bands

The *Bands* window determines the number of frequency bands used by the EVOC20 FB. It ranges from **5** to **20**.

Increasing the number of bands also increases the processor overhead.

High/Low Frequency

The blue bar shown just beneath the **emagic** logo is a multi-part control which is used to determine the lowest and highest frequencies allowed to “pass” by the filter. Frequencies which fall outside these boundaries will be cut. All filter bands are distributed evenly across the range defined by the **High/Low Frequency** values.

- To adjust the low frequency value, simply click-hold on the silver slider to the left of the blue bar, and drag to the right (or left). The value range is **75—750 Hz**.
- To adjust the high frequency value, simply click-hold on the silver slider to the right of the blue bar, and drag to the left (or right). The value range is **800—8000 Hz**.
- To adjust both sliders simultaneously, click on the area between the slider halves (directly on the blue bar) and drag to the left or right.
- You can make changes to the **High/Low** frequency values directly by using your mouse as a slider on the numerical entries—**80** and **8000Hz** in the diagram.

Lowest/Highest

These parameters can be found in the two small “windows” on either side of the Formant Filter window. These switches determine whether the lowest and the highest filter bands are band pass filters (just like all the bands between them), or whether they act as low pass/high pass filters, respectively. Click once on them to switch between the two curve shapes available.

- In the **Band Pass** setting, the frequencies below/above the lowest/highest bands are ignored.

- In the **High Pass** or **Low Pass** setting, all frequencies below the lowest (or above the highest) bands will also be treated.

Slope

The flip menu **Slope** determines the amount of filter slope applied to all filters of both filter banks. Choices are **1** (filter attenuation of 6 dB per octave) and **2** (filter attenuation of 12 dB per octave): **1** sounds softer, **2** sounds tighter.

Boost A/B Controls

The **Boost A** and **Boost B** knobs allow an increase or cut in the overall gain of the A and B filter banks. Their range is **±20 dB**. To adjust, click-hold and drag up or down with the mouse.

i You will need these controls, as the filter bank achieves its sounds by turning down the level of one or more filter bands. To make up for the resulting energy loss, use **Boost**.

i **Boost** is also quite handy to adjust the levels of both filter banks to each other, so that using **Fade A/B** (see below) leads only to a sound color change, but not to a level change.

Fade AB Control

The **Fade AB** crossfades between the A and B filter bank. At its extreme top or bottom position, you will only hear one of the filter banks.

Formant Shift

Moves the position of all bands in both filter banks up and down the frequency range.

i You can jump directly to the values **–0.5**, **–1**, **0**, **+0.5** and **+1** by clicking on their numbers.

Resonance

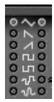
Resonance is responsible for the basic sonic character of both filter banks: low settings give it a soft character, high settings will lead to a more snarrling, sharp character. Increasing the value for **Resonance** emphasizes the middle frequency of each frequency band.

Modulation Parameters

The Modulation (LFO) area controls the **Formant Shift** and **Fade A/B** parameters of the EVOC 20 FB. It allows synchronous/non-synchronous modulation in bar, beat (triplet) or free values.

- *LFO Shift*, on the left-hand side, controls **Shift** modulation of the filter bands.
- *LFO Fade* controls the **Fade A/B** parameter.

Wave



These two switches allow the selection of the waveform type used by *LFO Shift* and *LFO Fade*. A selection of Triangle, falling and rising Sawtooth, Square up and down around zero (bipolar), Square up from zero (unipolar), a random stepped waveform (S&H), and a smoothed random waveform is available for each LFO.

Intensity

The **Intensity** sliders control the amount of **Formant Shift** and **Fade** modulation by the respective LFO's.

Rate Knobs

These knobs determine the speed of the modulation. Values to the left of the center positions are synchronized with the sequencer's tempo and include bar values, triplet values and more. Values to the right of the center positions are non-synchronous and displayed in Hertz (cycles per second).

The **Shift** and **Fade** LFO modulations are the keys to the most extraordinary sounds of the EVOC 20 FB: Make sure to set up totally different or complementary filter curves in both filter banks. Use rhythmic material like a drumloop as an input signal. Set up tempo-synchronized modulations—with different **Rates** for each LFO—for **Shift** and

Fade. And then try a tempo-synchronized Tape Delay after the EVOC 20 FB. You will end up with unique rhythms.

Output Section

Overdrive

This switch enables/disables the *Overdrive* circuit of the EVOC 20.

To actually hear the Overdrive effect, you may have to boost the level of one or both filter banks.

Level

The **Level** slider controls the level of the EVOC 20's output signal.

Stereo Mode

This flip menu determines the input/output mode of the EVOC 20 *Formant* filter bank. Choices are: **m/s**—mono input to *stereo* output and **s/s**—*stereo* input to *stereo* output.

- The **Stereo Mode** should be set to **m/s** if the signal going into the EVOC 20 FB is monophonic, for example a mono audio track.
- Stereo/stereo *(s/s)* is the preferred setting for stereo input signals. In this case, the stereo signal is processed by separate filter banks for the left and right channels. When using the **m/s** Mode on stereo input signals, the stereo signal is first summed to mono before it is passed on to the filter banks.

Stereo Width

Stereo Width distributes the output signals of the filter bands in the stereo field.

- In the left position, the output of all bands are centered.
- In the center position, the output of all bands ascends from left to right.

- In the right position, the bands are output evenly on the left and the right channel.

The stereo/stereo mode (**s/s**) uses one A/B filter bank per channel. The positioning of the frequency bands correspond to that described above, but the bands of each filter bank ascend in opposing directions, from left to right.

9 Tips for Better Speech Intelligibility

The “classic” vocoder effect is very demanding, with regard to the quality of *both* the *analysis* and the *synthesis* signal. Furthermore, the vocoder parameters need to be set carefully. Following are some tips on both topics.

The vocoder, in a way, always generates the “intersection point” of the analysis and synthesis signals. To explain: If there’s no treble portion in the *analysis* signal, the resulting vocoder output will also lack treble. This is also the case when the *synthesis* signal features a lot of high frequency content. This is true of each frequency band. As such, the vocoder demands a stable level in *all* frequency bands from *both* input signals, in order to obtain the best results. This leads to the first tip:

Compressing the Side Chain

The less the level changes, the better the intelligibility of the vocoder. We therefore recommend that compression be used in most cases.

Enhancing High Frequency Energy

The frequency spectrum should be dense and rich in the high frequency portion—of both the *analysis* and *synthesis* signals. If the side chain (analysis) signal consists of vocals or speech, a simple shelving filter should be sufficient. It doesn’t require much processing power, and efficiently boosts the high-mid and treble range, which is so important for speech intelligibility.

If the *synthesis* signal lacks treble energy, it can be generated with a distortion effect. The overdrive plug-in is perfect for this purpose.

The EVOC20 TO, used as a classic vocoder, with track 1 delivering the synthesis signal, and track 2 playing back the analysis (side chain) signal. On track 1, an overdrive plug-in generates additional harmonics in the treble range. The side chain signal is compressed, its treble is boosted by a shelving filter, and it is gated. The appearance of the windows looks a little different in Logic for Windows.

Avoiding Sonic Artifacts

A common problem with vocoder sounds are sudden signal interruptions (ripping, breaking sounds) and rapidly triggered noises, during speech pauses.

Release Parameter in the Analysis Section

The **Release** parameter defines the speed a given synthesis frequency band can decrease in level, if the signal level of the respective analysis band decreases abruptly. The sound is smoother when the band levels decrease slowly. To achieve this smoother character, use higher **Release** values in the analysis section of the interface. Longer release times result in a "washy" sound.

Short **Attack** values are no problem, in fact, they are even desirable for a fast reaction of the vocoder to impulse signals.

Gating Background Noises in the Side Chain

If the side chain signal is compressed, as recommended, the level of breath, rumble and background noises will rise. These background noises can cause the vocoder bands to open, but this is normally not intended. In order to eliminate these noises, it's therefore a good idea to employ a noise gate before compression and treble boosting. If the side chain signal is gated appropriately, you may find that you want to reduce the analysis **Release** value.

When gating speech and vocals, the **Hysteresis** parameter is important. **Threshold** defines the level, above which the gate will open. **Hysteresis** defines a lower threshold level, under which the gate will close. The value is relative to the **Threshold** level. The graphic shows a **Threshold** setting, which is well-adapted to compressed speech. Unwanted triggering by low or high frequency noise is avoided by the

noise gates' dedicated sidechain filters. The **Hold**, **Release** and **Hysteresis** values shown are well-adapted to "typical" level envelopes of most vocal and speech signals.

Treatment of the side chain signal (speech) with Compressor, Shelving Filter and Gate. The Silver Compressor, Silver Gate or another EQ are well-suited for these purposes.

Achieving the Best Analysis and Synthesis Signals

For a good speech intelligibility, please keep these points in mind:

- The spectra of the analysis and synthesis signals should overlap almost completely. Low male voices with synthesis signals in the treble range do not work well.
- The synthesis signal must be constantly sustained, without breaks. The track should be played legato, as breaks in the synthesis signal will stop the vocoders output. Alternatively, in the EVOC20 PS, the **Release** parameter of the synthesis signal (not to be confused with the **Release** time of the analysis section) can be set to a longer time. Nice effects can also be achieved by the use of a reverberation signal as a synthesis signal. Note that the two latter methods can lead to harmonic overlaps.
- Do not overdrive the vocoder. This can happen easily, and distortion will occur. Lower the **Gain** of the compressor in the side chain track, to avoid this problem.
- Pronounce the speech well, if the recording is to be used as an analysis signal. Spoken words, with a relatively low pitch, work better than sung vocals—even if the creation of vocoder choirs is your goal! Pronounce consonants well.

A nice example is the "rolled R" of "We are the Robots", by Kraftwerk, a classic vocoder track. This pronunciation was specifically made to cater to the demands of the vocoder.

Feel free to do what you like when setting the formant parameters. The intelligibility of speech is surprisingly little affected by shifting, stretching or compressing the formants. Even the number of frequency bands used has a minimal impact on the quality of the intelligibility. The reason for this is our ability to intuitively differentiate the voices of children, women and men, whose skulls and throats vary vastly by nature. Such physical differences cause variations in the formants which make up their voices. Our perception (recognition) of

speech is based on an analysis of the *relationships* between these formants. In the EVOC 20, these stay intact, even when extreme formant settings are used.

10 Vocoder History

You may think of the Vocoder as being a decidedly “futuristic” device with a fairly short history. If so, it may surprise you to learn that the “Voder” and “Vocoder” date back to 1939 and 1940, respectively.

Homer Dudley, a research physicist at Bell Laboratories, New Jersey USA developed the Voice Operated reCOrDER as a research machine. It was originally designed to test compression schemes for the secure transmission of voice signals over copper phone lines.

It was a composite device consisting of an analyzer and an artificial voice synthesizer. These were the:

- Parallel Bandpass Vocoder—speech analyzer and resynthesizer, invented in 1940.
- The Voder speech synthesizer—a voice model played by a human operator, invented in 1939. This valve-driven machine had two keyboards, buttons to recreate consonants, a pedal for oscillator frequency control, and a wrist-bar to switch vowel sounds on and off.

The analyzer detected the energy levels of successive sound samples measured over the entire audio frequency spectrum via a series of narrow band filters. The results of which could be viewed graphically as functions of frequency against time.

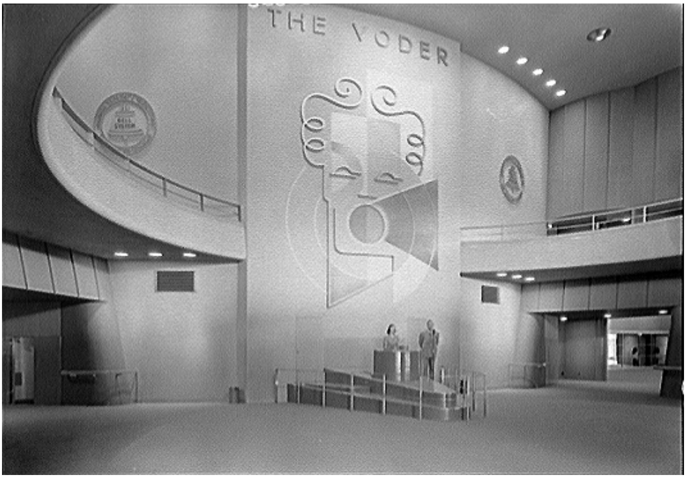
The synthesizer reversed the process by scanning the data from the analyzer and supplying the results to a number of analytical filters hooked up to a noise generator. This combination produced sounds.

The Voder was demonstrated at the 1939 World Fair, where it caused quite a stir.

In World War II, the Vocoder (now called VOice enCODER) proved to be of crucial importance, scrambling the transoceanic conversations between Winston Churchill and Franklin Delano Roosevelt.

Werner Meyer-Eppler, the director of Phonetics at Bonn University, recognized the relevance of the machines to electronic music after Dudley visited the University in 1948. Meyer-Eppler used the vocoder as a basis for his future writings which, in turn, became the inspiration for the German "Elektronische Musik" movement.

In the 1950's, a handful of recordings ensued.



The "Voder" at the 1939 World Fair.

In 1960, the Siemens Synthesizer was developed in Munich. Among it's many oscillators and filters, it included a valve-based vocoding circuit.

In 1967, a company called Sylvania created a number of digital machines that used time-based analysis of input signals, rather than band-pass filter analysis.

In 1971, after studying Duley's unit, Bob Moog and Wendy Carlos modified a number of synthesizer modules to create their own vocoder for the *Clockwork Orange* sound track.

Peter Zinovieff's company EMS in London developed a standalone—and altogether more portable—vocoder. EMS are

probably best known for the “Synthi AKS” and VCS3 synthesizers. The EMS “Studio Vocoder” vocoder was the world’s first commercially available machine, released in 1976. It was later renamed the “EMS 5000”. Among it’s users were Stevie Wonder and Kraftwerk. Stockhausen, the German “Elektronische Musik” pioneer, also used an EMS vocoder.

Sennheiser released the “VMS 201” in 1977, and EMS released the “EMS 2000”, which was a cut-down version of it’s older sibling.

1978 saw the beginning of “mainstream” vocoder use, riding on the back of popularity created through the music of Herbie Hancock, Kraftwerk and a handful of other artists. Among the manufacturers who jumped into vocoder production at this time are: Synton/Bode, Electro-Harmonix and Korg, with the VC-10.

In 1979, Roland released the VP 330 ensemble/vocoder keyboard.

The late 70’s and early 80’s were the heyday of the vocoder. Artists who used them included: ELO, Pink Floyd, Eurythmics, Tangerine Dream, Telex, Bowie, Kate Bush and many more.

On the production side, vocoders could (and can still) be picked up cheaply in the form of kits from electronics stores.

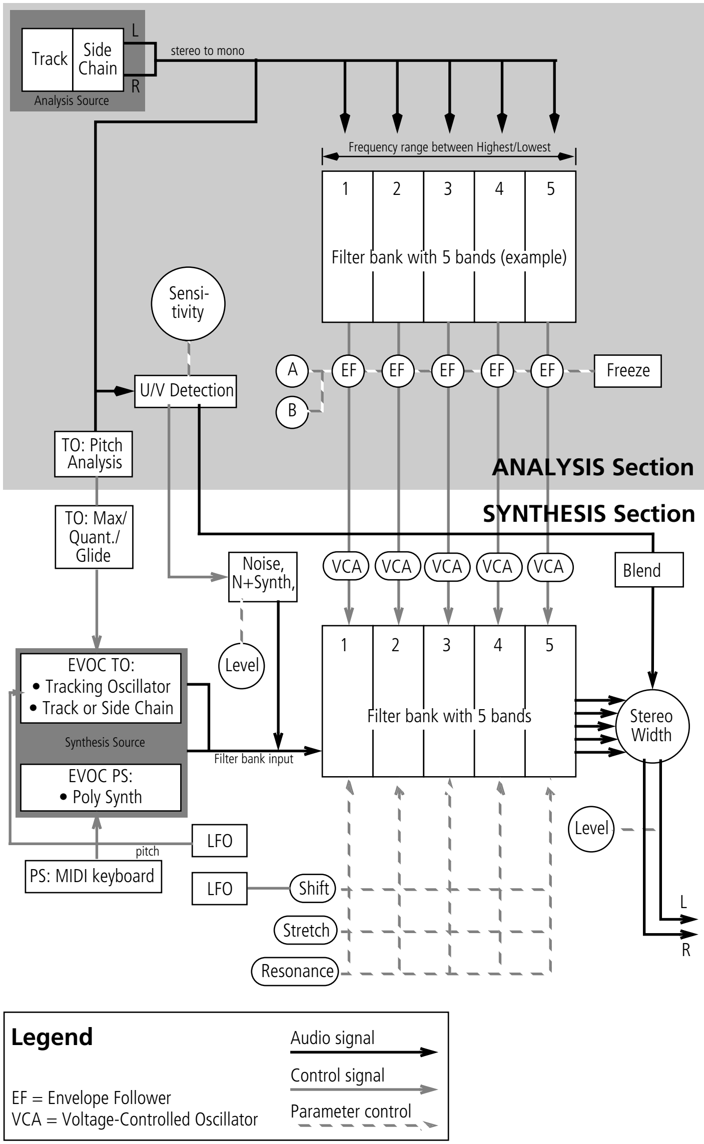
From 1980 through to the present, EMS in the UK, Synton in Holland and PAiA in the USA were, and remain, the main flyers of the vocoding flag.

In 1996, Doepfer in Germany and “Music and More” joined the vocoder-producing fraternity.

Throughout the 1990’s, a number of standalone software-based vocoders have appeared.

As you can see, the history of the vocoder is long and varied, and with 2002 seeing the birth of the EVOC 20 as a Logic plug-in, we’re sure that the vocoder will be with us for a while to come.

11 Block Diagram



12 MIDI Controllers Received

The following tables show the CC numbers used when the following MIDI preference is active: **Options** > **Settings** > **MIDI Options** > **(Version 4.x behavior)**.

EVOC20 PS

Sidechain Analysis	Attack	CC #77
	Release	CC #78
	Freeze	CC #76
Keyboard	Mode	CC #89
	Voices	CC #83
	Unison	CC #108
Synthesis	Oscillator Mode	CC #88
	Osc 1 Octave	CC #90
	Osc 1 Waveform	CC #94
	Osc 2 Waveform	CC #95
	Osc 2 Semitone	CC #93
	Detune/Ratio Fine	CC #91
	Osc Mix/FM Intensity	CC #100
Noise	Level	CC #99
	Color	CC #98
Analog	Tune	CC #84
	Intensity	CC #85
Glide	Time	CC #87
	Bend Range	CC #86
Oscillator Filter	Cutoff	CC #96
	Resonance	CC #97
Osc. Envelope	Attack	CC #106
	Release	CC #107

Formant Filter	FF Low Freq	CC #65
	FF Hi Freq	CC #66
	Formant Shift	CC #67
		CC #71
	FF Resonance	CC #72
	FF Low/Bandpass Select	CC #73
	FF High/Bandpass Select	CC #74
LFO 1 (Shift)	LFO 1 Rate	CC #68
	LFO 1 Waveform Select	CC #69
	LFO 1 Intensity	CC #70
LFO 2 (Pitch)	LFO 1 Rate	CC #102
	LFO 1 Waveform Select	CC #103
	LFO 1 Low Intensity	CC #104
	LFO 1 High Intensity	CC #105
U/V Detection	Sensitivity	CC # 81
	Mode	CC # 80
	Level	CC # 82
Output	Signal Out Select	CC #109
	Ensemble	CC #112
	Level	CC #110
	Mono/Stereo Select	CC #111
	Stereo Width	CC #79

EVOC20 TO

Sidechain Analysis	Attack	CC #79
	Release	CC #80
	Freeze	CC #78
Synthesis	Input Mode	CC #65
	Bands	CC #66
	FM Ratio	CC #86
	FM Intensity	CC #87
	Coarse Tune	CC #85
	Fine Tune	CC #105
Pitch Quantize	Strength	CC #90
	Root Scale Key	CC #103
	Root Scale Presets	CC #104
	Max Track	CC #88
	Glide Time	CC #89
Formant Filter	FF Low Freq	CC #67
	FF Hi Freq	CC #68
	Formant Shift	CC #69
	Formant Stretch	CC #74
	FF Resonance	CC #75
	FF Low/Bandpass Select	CC #76
	FF High/Bandpass Select	CC #77
LFO	LFO Rate	CC #70
	LFO Waveform Select	CC #71
	Shift Intensity	CC #72
	Pitch Intensity	CC #73
U/V Detection	Sensitivity	CC # 83
	Mode	CC # 82
	Level	CC # 84

Output	Signal Out Select	CC #106
	Level	CC #107
	Mono/Stereo Select	CC #108
	Stereo Width	CC #81
Root/Scale kyb.	C	CC #91
	C#	CC #92
	D	CC #93
	D#	CC #94
	E	CC #95
	F	CC #96
	F#	CC #97
	G	CC #98
	G#	CC #99
	A	CC #100
	A#	CC #101
	B	CC #102

EVOC20 FB

Filter Bands	Band Level	Bank A	Bank B
	Band 1	CC #64	CC #96
	Band 2	CC #65	CC #97
	Band 3	CC #66	CC #98
	Band 4	CC #67	CC #99
	Band 5	CC #68	CC #100
	Band 6	CC #69	CC #101
	Band 7	CC #70	CC #102
	Band 8	CC #71	CC #103
	Band 9	CC #72	CC #104
	Band 10	CC #73	CC #105
	Band 11	CC #74	CC #106
	Band 12	CC #75	CC #107
	Band 13	CC #76	CC #108
	Band 14	CC #77	CC #109
	Band 15	CC #78	CC #110
	Band 16	CC #79	CC #111
	Band 17	CC #80	CC #112
	Band 18	CC #81	CC #113
	Band 19	CC #82	CC #114
	Band 20	CC #83	CC #115
Boost	Bank A	CC #84	
	Bank B	CC #116	
Fade A/B		CC #117	

Formant Filter	Bands	CC #85	
	FF Low Freq	CC #88	
	FF Hi Freq	CC #89	
	Formant Shift	CC #90	
	FF Resonance	CC #91	
	Slope	CC #92	
	FF Low/Bandpass Select	CC #119	
	FF High/Bandpass Select	CC #120	
LFO 1 (Shift)	LFO 1 Rate	CC #93	
	LFO 1 Waveform Select	CC #94	
	LFO 1 Intensity	CC #95	
LFO 2 (Fade)	LFO 2 Rate	CC #121	
	LFO 2 Waveform Select	CC #122	
	LFO 2 Intensity	CC #123	
Output	Level	CC #124	
	Mono/Stereo Select	CC #87	
	Stereo Width	CC #86	

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